Consultative Committee for Space Data Systems

REPORT CONCERNING SPACE DATA SYSTEM STANDARDS

ADVANCED ORBITING SYSTEMS, NETWORKS AND DATA LINKS:

AUDIO, VIDEO, AND STILL-IMAGE COMMUNICATIONS SERVICES

CCSDS 704.1-G-3

GREEN BOOK

May 1994

AUTHORITY

Issue:	Green Book, Issue 3
Date:	May 1994
Location:	Villafranca, Spain

This document has been approved for publication by the Management Council of the Consultative Committee for Space Data Systems (CCSDS) and reflects the consensus of technical panel experts from CCSDS Member Agencies. The procedure for review and authorization of CCSDS Reports is detailed in reference [1].

This document is published and maintained by:

CCSDS Secretariat
Program Integration Division (Code OI)
National Aeronautics and Space Administration
Washington, DC 20546, USA

FOREWORD

Through the process of normal evolution, it is expected that expansion, deletion, or modification to this document may occur. This document is therefore subject to CCSDS document management and change control procedures which are defined in Reference [1].

At time of publication, the active Member and Observer Agencies of the CCSDS were

<u>Member Agencies</u>

- British National Space Centre (BNSC)/United Kingdom.
- Canadian Space Agency (CSA)/Canada.
- Central Research Institute of Machine Building (TsNIIMash)/Russian Federation.
- Centre National d'Etudes Spatiales (CNES)/France.
- Deutsche Forschungsanstalt für Luft- und Raumfahrt e.V. (DLR)/Germany.
- European Space Agency (ESA)/Europe.
- Instituto Nacional de Pesquisas Espaciais (INPE)/Brazil.
- National Aeronautics and Space Administration (NASA HQ)/USA.
- National Space Development Agency of Japan (NASDA)/Japan.

<u>Observer Agencies</u>

- Australian Space Office (ASO)/Australia.
- Austrian Space Agency (ASA)/Austria.
- Belgian Science Policy Office (SPO)/Belgium.
- Centro Tecnico Aeroespacial (CTA)/Brazil.
- Chinese Academy of Space Technology (CAST)/China.
- Communications Research Laboratory (CRL)/Japan.
- Danish Space Research Institute (DSRI)/Denmark.
- European Organization for the Exploitation of Meteorological Satellites (EUMETSAT)/Europe.
- European Telecommunications Satellite Organization (EUTELSAT)/Europe.
- Hellenic National Space Committee (HNSC)/Greece.
- Indian Space Research Organization (ISRO)/India.
- Industry Canada/Communications Research Center (CRC)/Canada.
- Institute of Space and Astronautical Science (ISAS)/Japan.
- Institute of Space Research (IKI)/Russian Federation.
- KFKI Research Institute for Particle & Nuclear Physics (KFKI)/Hungary.
- MIKOMTEK: CSIR (CSIR)/Republic of South Africa.
- Ministry of Communications (MOC)/Israel.
- National Oceanic & Atmospheric Administration (NOAA)/USA.
- Swedish Space Corporation (SSC)/Sweden.
- United States Geological Survey (USGS)/USA.

DOCUMENT CONTROL

ISSUE	TITLE	DATE	STATUS/ REMARKS
Green Book, Issue 1	CCSDS 704.0-G-1, Report concerning Space Data System Standards: Audio, Video and Still-Image Communication Services.	October 1992	Superseded
Green Book, Issue 2	CCSDS 704.0-G-2, Report concerning Space Data System Standards: Audio, Video and Still-Image Communication Services.	July 1993	Superseded
Green Book, Issue 3	CCSDS 704.1-G-3, Report concerning Space Data System Standards: Advanced Orbiting Systems, Networks and Data Links — Audio, Video and Still-Image Communication Services.	May 1994	Current Issue

CONTENTS

CONTENTS (continued)

CONTENTS (continued)

REFERENCES

[1] *Procedures Manual for the Consultative Committee for Space Data Systems*. CCSDS A00.0-Y-6. Yellow Book. Issue 6. Washington, D.C.: CCSDS, May 1994 or later issue.

[2] *Advanced Orbiting Systems, Networks and Data Links: Architectural Specification*. Recommendation for Space Data Systems Standards, CCSDS 701.0-B-2. Blue Book. Issue 2. Washington, D.C.: CCSDS, November 1992 or later issue.

[3] *Advanced Orbiting Systems, Networks and Data Links: Audio, Video, and Still-Image Communications Services*. Recommendation for Space Data Systems Standards, CCSDS 704.0-B-1. Blue Book. Issue 1. Washington, D.C.: CCSDS, May 1994 or later issue.

[4] *CCITT Blue Book*. Geneva: CCITT, 1989.

[5] *Telecommunications: Analog to Digital Conversion of Voice by 2,400 Bit/Second Linear Predictive Coding*. FED-STD-1015. Prepared by National Communications System Office of Technology and Standards. Washington, D.C.: GSA, November 1984.

[6] *Telecommunications: Analog to Digital Conversion of Radio Voice by 4,800 Bit/Second Code Excited Linear Prediction (CELP)*. FED-STD-1016. Prepared by National Communications System Office of Technology and Standards. Washington, D.C.: GSA, November 1984.

[7] *Broadcasting Service (Television)*. Vol. XI-1 of *Recommendations of the CCIR, 1990 (Also Resolutions and Opinions)*. Geneva: CCIR, 1990.

[8] *Information Technology—Digital Compression and Coding of Continuous Tone Still Images*. ISO 10918-1. Geneva: ISO, 1992.

[9] *Advanced Orbiting Systems, Networks and Data Links: Summary of Concept, Rationale and Performance*. Report Concerning Space Data Systems Standards, CCSDS 700.0-G-3. Green Book. Issue 3. Washington, D.C.: CCSDS, November 1992 or later issue.

[10] R. Carper. *Analysis of Video and Audio Data Transport*. Greenbelt, MD: GSFC, April 1989.

1 INTRODUCTION

1.1 PURPOSE OF THIS DOCUMENT

The purpose of this Report Concerning *Audio, Video, and Still-Image Communications Services* is to provide guidelines for the implementation of Agency-specific and CCSDS-standardized audio, video, and still-image services on the base of the Advanced Orbiting Systems (AOS) defined by CCSDS (see reference [2]).

In parallel, a Recommendation for *Audio, Video, and Still-Image Communications Services* (reference [3]) defines a set of standardized CCSDS audio, video, and still-image services for cross support between Agencies. From among the potential applications surveyed, those services of interest to most participating Agencies in CCSDS are identified.

2 AUDIO END-TO-END TRANSPORT MODEL

This section defines, from the user point of view, the functional and operational characteristics of the different audio services envisaged for space applications.

2.1 AUDIO LINK

The audio services envisaged here may be characterized by a limited number of attributes. These attributes are:

– audio signal quality;

– audio end-to-end delay;

– optional audio service attributes.

The following discussion of the above-mentioned audio link attributes is limited to aspects related to the typology of services in the CCSDS principal network (see reference [2]). Additional quality criteria, not discussed here, and assessment procedures for audio quality can be found in the CCITT recommendations for ground services (reference [4]).

2.1.1 Audio Signal Quality

For the scale of requirements in relation to the signal quality of audio space application, several CCSDS-recommended audio services are provided. These audio services cover the requirements ranging from good intelligibility for routine audio communication (also on a low data-rate audio channel) up to high fidelity (hi-fi) for entertainment-quality audio communication.

Parameters for the audio signal quality of the CCSDS-recommended audio services are:

– audio signal bandwidth;

– noise artifacts;

– audio data reliability;

– isochronism of audio data.

2.1.1.1 Audio Signal Bandwidth

The end-to-end audio signal transport shall be able to overtake and maintain an analog audio bandwidth up to 3.4 kHz for “Low Rate Audio” and “Operational Audio” services, up to 7 kHz for the “Enhanced Audio” service, and up to 15 kHz (stereo is here optional) for the “Hi-fi Audio” service.

2.1.1.2 Noise Artifacts

Factors contributing to the noise artifacts of an audio signal at the receiver side are

– existing noise on the incoming analog audio signal (at transmitter side);

– steady noise of quantization due to PCM-coding; and

– additional noise due to sporadic errors on the digital transmission link (space ↔ ground).

The validation of the noise contribution to the signal due to PCM coding has to be handled in an analog way just as for audio signals on the ground-networks, e.g., according to the CCITT recommendation G.711 or J.41/J.42 (reference [4]). The effect of the sporadic errors on the digital transmission link are described in more detail in 2.1.1.3.

2.1.1.3 Audio Data Reliability

The reliability criterion defines the behavior of the audio signal in case of errors on the digital transmission link. The applied CCSDS Principal Network (CPN) Grade of Service (see reference [2]) for audio data transmission has a strong influence on the data reliability of an audio channel.

A grade-3 CPN service provides no error correction to the audio data within a Service Data Unit (SDU) of a Virtual Channel Data Unit (VCDU). A Bit Error Rate (BER) of about $1x10^{-5}$ has to be considered (for example, on a 64 kb/s audio channel there is a statistical probability of one erroneous octet occurring every 1.5 seconds). But the bit errors mostly occur within a burst error distribution, so noticeable disturbance might be present every few seconds. This means the usability of grade-3 CPN services is limited.

A grade-2 CPN service provides a Reed-Solomon (R-S) error correction to the audio data within the SDU of the Coded VCDU (CVCDU). Because of this, a maximum BER of $1x10^{-7}$ is achievable, and normally a BER of better than $1x10^{-20}$ can be expected. So a grade-2 CPN service delivers an audio data stream with a very high probability of containing no errors (quasi-error free).

The probability for the loss of a whole VCDU/CVCDU because of misidentification of the VCDU/CVCDU header is a value of about $2.2x10^{-5}$ for grade-3 CPN services and, as a maximum, $1x10^{-20}$ for grade-2 CPN services.

In the case of a misidentified VCDU/CVCDU header, the data stream of the concerned audio channel is interrupted, and an artificial gap within the audio data stream occurs. The VCDU/CVCDU with the next audio data is used for continuation.

Detailed information of the error performance of international digital connections (based on ISDN) is given in CCITT Rec. G.821 (reference [4]).

2.1.1.4 Audio Data Isochronism

Isochronous transmission is considered to be a fundamental requirement for an end-to-end audio transport service. The timing interval between successive samples must be preserved or reconstructed. For transmission of audio data via CPN services, the isochronism can be reconstructed if an asynchronous CPN/Space Link Subnetwork (SLS) service provides the Audio Data Packets within the reconstruction buffer time constraints.

2.1.2 Audio Signal Delay

In practical implementations of space audio transmissions, the round-trip delay for full-duplex communication must be considered to be the result of a trade-off under the following aspects:

– given application requirements (e.g., operational audio in mission-critical phases or interactive telescience operation, where a short reaction time is necessary);

– the contributions of the signal travel time, which may already approximate the CCITT reference of 600 ms for duplex communication;

– the ability of trained people involved in the operations to cope with round-trip delays higher than the CCITT reference (therefore in reference [3] an end-to-end round-trip delay of 900 ms is given as “optimally maintained”).

More detailed information on the occurrence of round-trip audio signal delay (composed of the end-to-end delay contributions of two unidirectional audio services, space to ground and ground to space), is given in 5.3 below.

Within certain limits the end-to-end delay of a unidirectional audio service can be influenced by the choice and configuration of the CPN/SLS service transmitting the audio data (see annex B), but a minimal delay is given because of the signal travel time between the spacecraft and the ground station (depending on the number of geostationary hops), the space link processing latency, and a minimal required buffer time.

2.1.3 Optional Audio Service Attributes

The optional attributes of the audio service are:

– time tagging;

– audio data compression.

2.1.3.1 Time Tagging

For the transmission of the time tagging signal, a separate audio channel (of the same kind of audio service and multiplexed with the normal audio channel), assigned as auxiliary data channel, shall be used. The octet number of the time code entries should be adapted to the size of one Audio Data Block (ADB).

2.1.3.2 Audio Data Compression

Compressed audio data can save bandwidth on the transmission link. The quality of compressed audio data depends mainly on the compression algorithm used and on the available data rate. The transmission of compressed audio data requires error protection (grade-2 CPN service) in order to avoid error propagation within the decompressed audio data octet sequence. Recommended methods for audio data compression (see reference [3]) are CCITT G.721, G.722, G.728 (reference [4]) and methods given by FED standards (references [5] and [6]).

2.2 AUDIO SIGNALING

The audio transport model is, in general, connection oriented. In the space environment preconfiguration, permanent connections and on-demand connections can be distinguished. Both kinds of connections imply a variable audio service configuration and a mechanism to control this.

The audio signaling has to provide the parameter for creation and for control of the audio services and the related connections as well as the information on how to interpret the data transmitted.

Three different methods of signaling could be used:

– out-band audio signaling;

– in-band audio signaling;

– audio CPN-management signaling.

2.2.1 Out-Band Audio Signaling

Out-band audio signaling can provide signal information to the audio service end user via a transmission path outside the audio channel. This is not presently required.

2.2.2 In-Band Audio Signaling

In-band audio signaling can provide signal information to the audio service end user within the concerned audio channel along with the normal audio service data. This is not presently required.

2.2.3 Audio CPN-Management Signaling

Audio signaling realized as a subset of the general CPN signaling provides signal information to the CPN-management units in the spacecraft and the concerned ground station. The signaling procedure of the CPN-signaling will be described in a separate CCSDS Blue Book.

The main information and parameters necessary for establishment, control, and termination of permanent or on-demand audio channels are:

– user address/access point;

– audio channel number;

– audio service used;

– data structure (compressed/uncompressed data);

– compression algorithm parameter, if necessary;

– time code insertion ON/OFF;

– type of CPN/SLS service used;(*)

– Virtual Channel (VC) number;(*)

– VCDU/CVCDU or Audio Data Packet (ADP) frequency;(*)

– m, the number of Audio Data Fields (ADFs) in an ADP;(*)

– n, the number of ADBs in an ADF;(*)

– p, the number of audio data octets per ADB.(*)

(*) For detailed information, see Section 5, “Audio transport over the CPN.”

3 VIDEO END-TO-END TRANSPORT MODEL

This section defines, from the user point of view, the functional and operational characteristics of the different video services envisaged for space applications.

3.1 VIDEO LINK

The video services envisaged here shall be characterized by a limited number of attributes. These attributes are:

– video sequence and image parameter;

– video data end-to-end delay;

– optional video service attributes.

The following discussion of the above-mentioned video link attributes is limited to aspects related to the typology of services in the CPN.

3.1.1 Video Sequence and Image Parameter

The intention regarding how video information is to be used provides the main distinction between classifications of different video services.

Video or image information is intended mainly for either visual use or quantitative use. Ground television networks are an examples of video systems designed to satisfy the requirements of visual use: they exploit the limitations of the human visual system for technical optimization.

For space science, image information has visual use as well as quantitative use, i.e., post-image processing and analysis. Particularly for these scientific applications, the “Still-Image” services (see section 4) are recommended.

The video sequence and image parameters of the CCSDS-recommended video services are:

– temporal parameter of the video sequence;

– spectral parameter;

– geometric parameter;

– radiometric parameter;

– video data reliability.

3.1.1.1 Temporal Parameter of the Video Sequence

Video information consists of a sequence of images considered to be basic samples of video-information flow with respect to time. These images may or may not be periodically sampled.

While periodic sampling is the rule in television systems, modern electronics allow practical solutions for still-image acquisition. Image sequences with a very low (below 1 per second) sampling period may, for all practical purposes, be assimilated to “still-images” (see section 5).

In video applications where periodic sampling contributes to faithful movement rendition, the original timing relationship between successive samples may be preserved along the video link or reconstructed at the receiving end.

The periodicity of the images in the currently envisaged video services for space applications may be 29.97 Hz, 14.99 Hz, 25 Hz, 12.5 Hz, or ≤5 Hz.

If a video service (e.g., V2, V3, V4) allows more than one frame rate, the equipment on the receiver side has to be adapted to the frame rate of the video data source. If, after receiving that data, a frame rate conversion is necessary to adapt the source video frame rate to the national common frame rate, the conversion method has to be chosen in accordance with the video quality requirements for the application.

3.1.1.2 Spectral Parameter

The spectral parameter defines the delivery of colored or monochrome video data. Monochrome video information consists only of the luminance signal Y. If the video service provides color, a component video signal consisting of the Y component and two additional chrominance components, Cr and Cb (as separate color-difference components), has to be transmitted within the video data stream. Another possibility for a color video service is to provide the three color components, R, G, and B.

3.1.1.3 Geometric Parameter

The images provided by video services consist of orthogonally arranged image elements (pels) organized in lines and rows. The spatial resolution gives the number of rows (pels per line) and the number of scanned lines within one image.

The spatial resolution spectrum of the envisaged video service for space application ranges from the Quarter Common Intermediate Format (QCIF, with 176*144 pels for Y and 88*72 pels for the chrominance components Cr and Cb) up to a resolution provided by a future HDTV video standard.

If a video service (e.g., V3, V4) allows more than one spatial frame definition, the equipment on the receiver side has to be adapted to the spatial frame definition of the video data source. If, after receiving that data, a spatial resolution conversion is necessary to adapt the source frame resolution to the national common frame resolution, the conversion method has to be chosen in accordance with the video quality requirements for the application.

3.1.1.4 Radiometric Parameter

The radiometric parameter defines the resolution of the sampled pel-signal-value (pixel value). The 8-bit representation of the video signal components Y, Cr, and Cb is defined in CCIR Recommendation 601 (reference [7]):

for Y: Black = 16
White = 235

for Cr,Cb: Zero color difference = 128
Peak color difference = 16 and 240

3.1.1.5 Video Data Reliability

The reliability criterion defines the behavior of the video signal in case of errors on the digital transmission link. The applied CPN Grade of Service has a strong influence on the video data reliability of the video channel.

A grade-3 CPN service provides no error correction to the video data within an SDU of a VCDU. A BER of about $1x10^{-5}$ has to be considered for a grade-3 CPN service data transmission, which is not acceptable for compressed video data and which will also diminish the quality of uncompressed video.

A grade-2 CPN service provides an R-S error correction to the video data within the SDU of the CVCDU. As a result, a maximum BER of $1x10^{-7}$ is achievable, and normally a BER of better than $1x10^{-20}$ can be expected. So a grade-2 CPN service delivers a video data stream with a very high probability of containing no errors (quasi-error free).

The probability for the loss of a whole VCDU/CVCDU due to misidentification of the VCDU/CVCDU header is a value of about $2.2x10^{-5}$ for grade-3 CPN services and, as a maximum, $1x10^{-20}$ for grade-2 CPN services.

Detailed information of the error performance of international digital connections (based on ISDN) is given in CCITT Rec. G.821 (reference [4]).

3.1.2 Video Signal Delay

Some implementations of space video services have to consider video communication in only one direction, where a video data end-to-end delay of several image times can (within certain limits) be accepted. For other applications with bidirectional video communication (composed of two unidirectional video links) possibly synchronized to audio signals (lip synchronization), a short round-trip-delay video signal delay is a strong requirement for the implementation of the video services. The delay can be influenced by the choice and configuration of the CPN/SLS service transmitting the video data, but a minimal delay is given because of the signal travel time between the spacecraft and the ground station (depending on the number of geostationary hops), the space link processing latency, and a minimal required buffer time. More detailed information about the signal delay can be found in 5.3 below or in Annex B (although this information is given for audio examples, it is valid in much the same way for video).

3.1.3 Optional Video Service Attributes

The optional attributes of a video service are:

- time tagging;
- video data compression.

3.1.3.1 Time Tagging

The transmission of the time tagging signal can be provided by insertion of time code octets into the image data headers of the video sequence.

3.1.3.2 Video Data Compression

Compressed video data can save bandwidth on the transmission link. The transmission of compressed video data requires error protection (grade-2 CPN service) in order to avoid error propagation within the decompressed video data. The image quality of compressed/ decompressed video data depends mainly on the compression algorithm used and on the available data rate. Several algorithms for video data compression are available, providing different grades of restitution quality of the video data after decompression. The video coding can be divided into 3 categories: ‘lossy coding’, ‘irrelevance coding’, ‘redundancy coding’, where

- ‘redundancy coding’ refers to a fully reversible coding and compression algorithm with no information loss;
- ‘irrelevance coding’ refers to a coding/compression algorithm that is not fully reversible, with which image data will be altered within the domain where only visually irrelevant information may be lost;
- ‘lossy coding’ refers to a coding/compression algorithm that is not fully reversible, with which the quality of the reconstructed images complies with a specified goal.

It is desirable to use standardized algorithms for video compression when the complexity of on-board equipment may be reduced. Such a standard will be for a ‘lossy coding’ (V2 Moderate Resolution Video): CCITT recommendation H.261 (codec for audio visual services) (see reference [4]). For the CCSDS Broadcast Resolution Video service ‘V3’, the lossy compression coding algorithm AD/CMTT is available.

3.2 VIDEO SIGNALING

The video transport model is connection oriented. As for audio preconfiguration, permanent connections and on-demand connections for video data transmission are possible. Both kinds of connections imply a variable video service configuration and a mechanism to control it.

The video signaling has to provide both the parameters for creation and for control of the video services and related connections as well as the information about how to interpret the data transmitted.

Three different methods of signaling could be used:

– out-band video signaling;

– in-band video signaling;

– video CPN-management signaling.

3.2.1 Out-Band Video Signaling

Out-band video signaling can provide signal information to the video service end user via a transmission path outside the video channel. This is not presently required.

In order to minimize the implementation complexity in case of the CCSDS Moderate Resolution Video service, 'V2', using the CCITT Recommendation H.261 video coding and compression algorithm (reference [4]), the out-band signaling features of the coder will not be adapted to CCSDS.

3.2.2 In-Band Video Signaling

In-band video signaling can provide signal information to the video service end user within the concerned video channel along with the normal video service data. This is not presently required.

3.2.3 Video CPN-Management Signaling

Video signaling realized as a subset of the general CPN signaling provides signal information to the CPN-management units in the spacecraft and the concerned ground station. The signaling procedure of the CPN-signaling will be described in a separate CCSDS Blue Book.

The main information and parameters necessary for establishment, control, and termination of permanent or on-demand video channels are:

– user address/access point;

– video channel number;

– video service used;

– time code insertion ON/OFF;

– data structure:

  • compressed/uncompressed video data;

  • embedded audio data;

– if necessary, compression algorithm parameter;

– type of CPN/SLS service used;[*]

– VC number;[*]

– VCDU/CVCDU or Video Data Field (VDF) frequency;[*]

– n, the number of Video Data Blocks (VDBs) in a VDF;[*]

– p, the number of octets per VDB.[*]

[*] For detailed information, see Section 6, “Video transport over CPN.”

4 STILL-IMAGE END-TO-END TRANSPORT MODEL

This section defines, from the user point of view, the functional and operational characteristics of the different still-image services envisaged for space applications.

4.1 STILL-IMAGE LINK

The still-image services envisaged here shall be characterized by a limited number of attributes. These attributes are:

– still-image parameters;

– optional still-image service attributes.

The following discussion of the above-mentioned still-image link attributes is limited to aspects related to the typology of services in the CCSDS principal network.

4.1.1 Still-Image Parameter

Still-image information is intended mainly for quantitative use (i.e., space science post-image processing and analysis) and sometimes also for visual use.

The image parameters of the CCSDS-recommended still-image services are:

– spectral parameter;

– geometric parameter;

– radiometric parameter;

– still-image data reliability.

4.1.1.1 Spectral Parameter

The spectral parameter defines the delivery of colored or monochrome still-image data. Monochrome still-image information consists only of the luminance signal, ‘Y’. If the still-image service provides color, a component image signal consisting of the Y component and two additional chrominance components, ‘Cr’ and ‘Cb’ (as separate color difference components), has to be transmitted within the still-image data stream. Another possibility for a color still-image service is to provide the three color components R, G, and B.

The envisaged still-image services are able to handle all kinds of color image signals including the possibility of only a Y-signal. Infrared and other kinds of spectral component images can be transmitted by interpreting them as single-component images (like Y-images).

4.1.1.2 Geometric Parameter

The images provided by still-image services consist of orthogonally arranged image elements (pels) organized in lines and rows. The spatial resolution gives the number of rows (pels per line) and the number of scanned lines within one image.

The envisaged still-image services can cope with each resolution desired.

4.1.1.3 Radiometric Parameter

The radiometric parameter defines the resolution of the sampled pel-signal-value (pixel value).

The envisaged still-image services can cope with 8-bit/pel representations using the CCSDS Still-Image Mode-3 service I3 (lossy coding) and up to 16-bit/pel, or unlimited bits/pel using the still-image services I2 or I1, respectively, with redundancy coding or no coding of the image data.

4.1.1.4 Still-Image Data Reliability

The reliability criterion defines the behavior of the still-image signal in case of errors on the digital transmission link. The applied CPN Grade of Service has a strong influence on the still-image data reliability.

A grade-3 CPN service provides no error correction to the still-image data within an SDU of a VCDU. A BER of about 1×10^{-5} has to be considered for a grade-3 CPN service data transmission, which is not acceptable for compressed still-image data and which will also diminish the quality of uncompressed still-images. In the case of quantitative-use still-image data, these distortions in the uncompressed data can be a disruptive factor.

A grade-2 CPN service provides an R-S error correction to the still-image data within the SDU of the CVCDU. Because of this, a maximum BER of 1×10^{-7} is achievable and normally a BER of better than 1×10^{-20} can be expected. So a grade-2 CPN service delivers a still-image data stream with a very high probability of containing no errors (quasi-error free).

Detailed information on the error performance of international digital connections (based on ISDN) is given in CCITT Recommendation G.821 (reference [4]).

4.1.2 Optional Still-Image Service Attributes

The optional attributes of a still-image service are:

- time tagging;
- still-image data compression.

4.1.2.1 Time Tagging

The transmission of the time tagging signal can be provided by insertion of the time code octet into the still-image data header.

4.1.2.2 Still-Image Data Compression

Compressed still-image data can save bandwidth on the transmission link. The transmission of compressed still-image data requires error protection (grade-2 CPN service) in order to avoid error propagation within the decompressed still-image data. The image quality of compressed/decompressed still-image data depends mainly on the compression algorithm used. Several algorithms for image compression are available providing different grades of restitution quality of the still-images after the decompression.

The reconstructed images, coded with a compression algorithm using ‘irrelevance’ or even ‘lossy’ coding, can comprise some pixel value distortions in reference to the original input image. These distortions, which are often visually not noticeable, can be influenced by parameters of the coding method/algorithm, so that only a very small loss of image information can be achieved (which could equal only the noise in the original image information). The compression ratio of such image coding methods can achieve a figure up to 30 or even higher.

Still images coded with ‘redundancy’ coding algorithms can be reconstructed to 100%; i.e., no image information will be lost. The compression ratio of such image coding methods is relative low (1:3).

It is desirable to use standardized algorithms for still-image data compression when the complexity of on-board equipment may be reduced. Such a standard mainly for ‘redundancy’ and ‘lossy’ coding will be the ISO/IEC CD 10918-1 (called JPEG) (reference [8]). The lossy coding mode of ISO/IEC CD 10918-1 is based on an Adaptive Discrete Cosine Transform (ADCT) method. The redundancy coding or lossless mode is based on a DPCM method.

4.2 STILL-IMAGE SIGNALING

No continuous data channel like audio or video channels has to be established for the still-image services, because the still-image services are designed and intended for delivering data of single images as isolated units with determined volume. The end-to-end connection management of the still-image services provides a variable still-image service configuration, and a mechanism is needed therefore to control it.

The still-image signaling has to provide the parameter for controlling of the CCSDS still-image service and the related CPN/SLS-service as well as the information on how to interpret the transmitted still-image data.

Three different methods of signaling could be used:

– out-band still-image signaling;

– in-band still-image signaling;

– still-image CPN-management signaling.

4.2.1 Out-Band Still-Image Signaling

Out-band still-image signaling can provide signal information to the still-image service end user via a transmission path outside the still-image service data. This is not presently required.

4.2.2 In-Band Still-Image Signaling

In-band still-image signaling can provide signal information to the still-image service end user within the concerned still-image service data along with the normal image data. This is not presently required.

4.2.3 Still-Image CPN-Management Signaling

Still-Image signaling realized as a subset of the general CPN signaling provides signal information to the CPN-management units in the spacecraft and the concerned ground station. The signaling procedure of the CPN-signaling will be described in a separate CCSDS Blue Book.

The main information and parameters necessary for service control are:

– Still-Image service used;

– data structure: compressed/uncompressed data;

– compression algorithm parameter, if necessary;

– time code insertion ON/OFF;

– type of CPN/SLS service used;(*)

– user address/access point;

– the 'Logical Data Path'.

(*) For detailed information: Section 7, "Still-Image transport over the CPN."

5 AUDIO TRANSPORT OVER THE CPN

This section defines the audio data handling for the audio data transmission via the CPN/SLS.

Audio data can only be transmitted unidirectionally over the SLS via specific audio data channels or as embedded audio data within video data (if the video coder or specific audio/video multiplexing devices [working for example in accordance to CCITT Rec. H221/H.222/H.242 Draft] provide this feature). In the case of embedded audio data transmission, the synchronization of audio and video data can be maintained within the accepted time shift tolerances in a very simple way.

In the case of separate audio data transmission, digital data streams containing multiplexed data from several audio channels (isochronous audio data octets or time tagging data) are created. One audio data stream can carry either one audio channel or several audio channels multiplexed together. In the latter case, all the audio channels must belong to the same audio service. Different audio services generate different audio data streams.

The structure of multiplexed audio data streams is defined in 5.1. The applicable CPN/SLS services for the audio data transmission are introduced in 5.2. An overview of signal delay and audio signal resynchronization is presented in 5.3. Some bidirectional audio communication aspects are given in 5.4.

5.1 AUDIO SERVICE DATA MULTIPLEXING MODEL

This subsection describes how to multiplex several audio channels (of the same service type) within a separate audio data stream (not as embedded audio data in video data).

Three different data structures are used:

– **Audio Data Block**. The Audio Data Block (ADB) data structure contains p contiguous octets of the same audio channel (see figure 5-1).

– **Audio Data Field**. The Audio Data Field (ADF) data structure contains n ADBs, n being the number of audio channels multiplexed in the audio data stream. The n ADBs are arranged sequentially from ADB#1 to ADB#n within the ADF.

– **Audio Data Packet**. The Audio Data Packet (ADP) data structure contains m ADFs. ADPs represent an optional multiplexing layer, which should be used only when the audio data stream is transmitted over the space link using the Insert service. In all other cases, the ADP multiplexing layer should be discarded.

It is required that the n audio channels multiplexed within the ADF belong to the same audio service. The values of n and p (and m when ADPs are used) should be optimized together, taking into account:

- the type and bit rate of the CPN/SLS service used over the space link;
- the audio channel bit rate;
- the required end-to-end delay for the audio data.

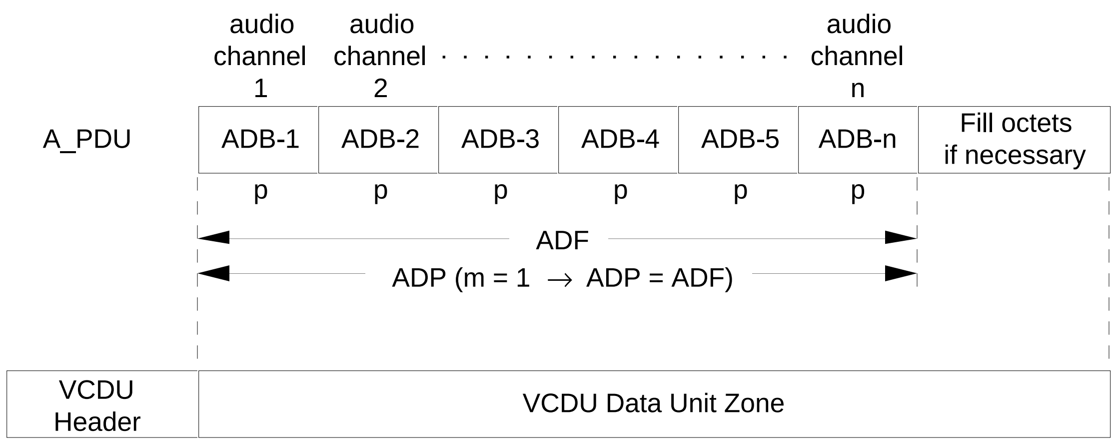

Figure 5-1: Audio Service Data Multiplexing Model

5.1.1 Audio Data Block

An ADB is composed of p contiguous octets from the same audio channel. In general, audio data is octet aligned at the output of the audio coder. If the audio data might not be octet aligned at the output of the audio encoder, the octet alignment should be artificially created. The octet alignment should be preserved throughout the transmission over the space link by aligning audio data octets with ADB octets.

The value of p has a considerable influence on the end-to-end audio signal delay (for detailed information about delay see 5.3). The value of p and the assignment of audio channels to ADB slots is not signaled in-line within the audio formats. This information is supposed to be transmitted separately from the audio data by the CPN management and signaling service.

The ADB is a fixed-length data structure of p octets. The p octets are all valid audio data. This implies that, for a given audio channel, ADBs are produced at a fixed rate of

$$F = (ABR/8*p) \text{ blocks/sec}$$

where

ABR = Audio Channel Bit Rate (fixed within the frequency shift tolerance).

The ADF is a multiplex of n ADBs (n being fixed = number of multiplexed audio channels). Therefore, ADFs are produced at a fixed rate equal to F.

In the case where a data driven-mode of access to the CPN/SLS transmission services (nonisochronous CPN/SLS service) is used, ADFs/ADBs are buffered and are transmitted asynchronously with respect to the fixed rate F. The CPN/SLS service used should guarantee that each ADF shall not be delayed more than a given specification.

In the case where an isochronous CPN/SLS service is used to transmit the audio data stream (e.g., Insert service), it is necessary to have a very good match between the ADF/ADB rate and the SDU (e.g., Insert data unit) rate. A very good match means:

if F = ADF/ADB rate
if F’= SDU rate

we should have: $(F-F')/F < 5.10^{-5}$; (in line with CCITT rec.711/721/722).

This figure guarantees that the number of lost octets in the transmission due to synchronization problems is sufficiently low, so that audio quality for A2/A3/A4 services is not impaired.

5.1.2 Audio Data Field

The ADF is a fixed-length data structure consisting of n ADBs, n being the number of multiplexed audio channels within the audio data stream. n*p has to be smaller than or equal to the size of the SDU of the applied CPN/SLS service. The n ADBs are arranged in sequence within the ADF.

All the ADBs within an ADF represent an audio channel of the same kind of audio service. The number of octets per ADB “p” has to be the same for all ADBs within the ADF. To prevent a jitter or delay of the incoming data out of the “n” multiplexed audio channels in the ADF, the same sampling clock should be used to generate the digital audio service data octets in all these channels.

The ADF will be inserted at the beginning of the SDU of the applied CPN/SLS service.

The value of n is not transmitted inline in the ADF structure. It is transmitted by the CPN management and signaling system.

5.1.3 Audio Data Packet

The ADP is a fixed-length data structure consisting of m ADFs. This extra multiplexing layer should be used only if transmitting the audio data stream over the space link using the Insert service. ADPs enable byte interleaving of audio channels, thus giving very regular access to the link to each of the audio channels. The delay encountered over the space link by each audio channel is exactly the same. When ADPs are used, the following parameters should be used:

– p = 1 octet (ADB-size);

– n = number of audio channels multiplexed;

– m = (Insert-size)/n.

The ADP will be inserted at the beginning of the SDU of the applied CPN/SLS service.

The value of m is not transmitted inline in the ADP structure. It is transmitted by the CPN/SLS management and signaling system.

5.2 CPN/SLS SERVICES FOR AUDIO

This subsection describes in detail the applicable CPN/SLS services to be used to transmit the data of the various audio services. At a given time in the mission, each active audio service generates its own audio data streams. Each audio data stream contains one or more audio channels. Each audio data stream will interface with a selected CPN/SLS service and be inserted in the form of ADFs or ADPs to its SDUs.

5.2.1 Audio on the Insert Service

Insert service is an isochronous service which uses a fixed slot within the Virtual Channel Data Unit (VCDU) to transmit data. Insert service should be used only for those audio data needing very low end-to-end delay, over a space link having a relatively low data rate (i.e, lower than 10 Mbps). With this service, access to the space link is guarantied every VCDU. That means that the latency encountered by the data being transmitted in the Insert SDU is minimal. On the contrary, Insert service, being a synchronous access service, requires that the ADFs or the ADPs are produced at the exact rate of the VCDU. This drives several constraints because the audio system (onboard or on the ground) has to be synchronized and locked with the space link multiplexing system (VCDU multiplex) or vice-versa. Therefore, Insert service for audio transmission should be used only when no other solution exists given the overall bit rate of the link and the delay requirement on the audio.

To further reduce the audio delay when using the Insert service, it is useful to do the following:

– Read out audio data contained in the Insert before R-S decoding is performed on the VCDU. This enables audio readout before the complete CVCDU is received. The extra protection given by the R-S code is in general not needed for audio data (which requires only 10^{-4} end to end).

– Use the ADP structure with the following parameters:

p = 1 (each ADB is one octet of an audio channel);
n = number of multiplexed audio channels;
m = soi/n (with soi = size of Insert SDU [in octets]).

With this system, the delay encountered by each audio channel is the same, because each channel has access to the link every n octets during the Insert sampling time. This system is equivalent to a byte interleave of the audio channels. It is not useful with services other than Insert.

5.2.2 Audio on the Virtual Channel Access Service

The sequence-preserving Virtual Channel Access service (VCA) can be used for audio data stream transmission. Formatted ADFs are inserted into successive VCDUs or CVCDUs of the VCA service. The structure of the ADFs/ADBs (i.e., values of p and n) is transmitted from sender side to receiver side by the CPN management and signaling system. This information is needed at both ends of the audio link to establish communication.

The ADFs of a given audio data stream will be transmitted in the VCA SDUs of a given VC.

Link access of the VCA SDU will be one of the following:

– Asynchronous, the audio VC having a high priority, thus guaranteeing that the jitter and delay encountered by audio will be kept relatively low. This mode of operation requires that the overall bit rate of the space link is relatively high.

– Synchronous, the audio VC being sampled periodically by the VC multiplexer. This implies that the VCDU repetition rate is linked to the ADF repetition rate by an integer ratio, which in turn implies that the audio system is synchronized with the space link transmission system. Therefore, this synchronous mode of operation should be avoided whenever possible.

The ADF should be inserted at the beginning of the VCA SDU (i.e., VCDU/CVCDU data field), starting from the first octet.

The length of the ADF submitted to the VCA service cannot be larger than the size of the VCA SDU (i.e., VCDU/CVCDU data field). If the length of the ADF is smaller than the size of the SDU, fill octets must be added after the ADF to fill in the SDU. Of course, the number of fill octets must be minimized to avoid losing bandwidth. The size of the ADF must be tailored to be as close as possible to the size of the VCA SDU (VCDU/CVCDU data field).

The VCA service should be the standard CPN service for audio service data transmission over the space link.

5.2.3 Audio on the Encapsulation Service

As an option the audio data transfer via Encapsulation service is allowed, if the data traverse only the Space Link Subnet.

The Encapsulation service is a sequence-preserving service for the transmission of variable-length, octet-aligned data in an asynchronous mode. For data transmission, the Encapsulation service takes a delimited user data unit and encapsulates it within CCSDS packets. These are multiplexed with other packets in the same packet channel and with packets from other packet channels into dedicated VCs.

The Encapsulation service is inherently asynchronous, although some time constraints may be placed on the bearer VC.

For more information see Annex C of reference [3].

5.2.4 Audio on the Path Service

As an option, the audio data transfer via Path service is allowed.

The Path service may be of interest for some audio sources which require flexibility of placement of the audio codecs and of data transmission routing in the onboard and ground networks. The Path service provides internetworking using CCSDS packets as the protocol data unit. As with the Encapsulation service, transport of variable-length delimited user data structures is supported.

Care should be taken in the use of this service because of the unpredictable delay characteristics of the underlying onboard/ground subnets and because of their influence on the data sequencing. The Path service does not provide guaranteed in-sequence data over subnets which may corrupt the sequencing. The audio application may therefore need to apply its own sequencing mechanism. The CCSDS packet does provide a 14-bit packet sequence count field which may be used in this context. Alternatively, the Path service may generate a “Data Loss Indicator” where sequencing is violated.

For more information see Annex C of reference [3].

5.2.5 Audio on the Bitstream Service

The Bitstream service is not recommended for transmitting audio over the space link, for the following reasons:

– Using the Bitstream service, there is no possibility of synchronizing the ADF data structure with the VCDU/CVCDU data structure that carries it over the space link. That means that there is no way, using this service, to know where ADF starts and ends within the Bitstream SDU. Therefore, it is impossible to extract the ADFs from the Bitstream SDU.

– Synchro field would have to be added at the beginning of the ADF format to be able to perform this extraction. Given the additional overhead, this solution has been discarded, and Bitstream service is not recommended for audio transmission.

5.2.6 CPN Grades of Service to be used for Audio Transmission

The CPN provides 3 Grades of Service, which give 3 different levels of error-detection and correction capability. The error-handling functions are based on the fixed-length VCDU/ CVCDU data structure transmitted over the space link. Grade of Service is a characteristic associated with a VC. Depending on the VC used to transmit a particular audio SDU over the space link, the error protection will be different.

The grade-3 CPN service provides no error check or recovery to the user data (audio data) within the VCDU. An audio data transmission on a grade-3 VC minimizes the required bandwidth for this audio VC (check bits of the R-S code are replaced by audio data), but the BER encountered by the audio data over the space link is higher. Therefore, grade-3 should be used for a particular audio service if it is compatible with the end-to-end BER specified for this service (see annex D of reference [3]). The specified value for each service guarantees that the quality of audio signal is not significantly degraded by the transmission errors.

The grade-2 CPN service provides an error-detection and correction function for each CVCDU based on a R-S code. In practice, this type of VC can be considered to provide error-free transmission over the space link. Using a grade-2 VC reduces the size of the VCDU data field due to check bits of error code. Therefore, the bandwidth needed to transmit the same audio data stream will be approximately 15% higher on a grade-2 VC compared to a grade-3 VC. As a result, it is recommended that a grade-2 VC be used for a given audio service data transmission only if grade-3 is not able to meet the end-to-end BER requirement of this audio service, or if, because of design constraints on the space link communication system, all the VCs are grade-2.

The grade-1 CPN service provides, in addition to the error-detection and correction function of the R-S code, an Automatic Repeat and Queuing (ARQ) procedure, which guarantees (with a very high probability) the completeness and the correctness of the transmitted CVCDUs.

Because of the retransmission protocol used, this service is not sequence preserving and is not acceptable for audio data. Therefore, grade-1 CPN service should not be used for audio.

In conclusion, only grade-2 and grade-3 sequence-preserving CPN services can be used for audio service data transmission, when using the VCA service. The choice between the two should be made taking into account the end-to-end BER requirement of each audio service as defined in Annex D of reference [3]. When using the Insert service to transmit audio, the Grade of Service of all the VCs on the space link is necessarily grade-2 (as specified in CCSDS AOS Blue Book, reference [2]). Nevertheless, if Insert service is used for audio to minimize the end-to-end delay, it is recommended that the audio Insert be extracted before R-S decoding (that is, before the complete CVCDU has been received, and as soon as the first octet of audio Insert is received). This minimizes audio delay, the drawback being that audio data do not benefit from the additional protection of the R-S code. Therefore, when using Insert for audio, it should be checked that the BER performance of the space link without R-S is compatible with the required end-to-end BER of the considered audio service. If not, R-S decoding should take place before Insert extraction, thus increasing audio end-to-end delay.

Detailed information about the CPN Grades of Service can be found in CCSDS AOS Blue and Green books (references [2] and [9]).

5.3 END-TO-END AUDIO SIGNAL DELAY

5.3.1 Delay Overview

Audio signals transmitted in space audio links using geostationary relay satellites will get a delay. The end-to-end delay consists of several delay contributions:

- the signal travel time on the link;
- time of transmit/receive buffering and restitution of isochronism;
- space link processing latency;
- possible data front-end processing (e.g., filtering);
- possible ground network delays.

Figure 5-2 gives an overview of the delay contributions for a unidirectional audio signal transmission:

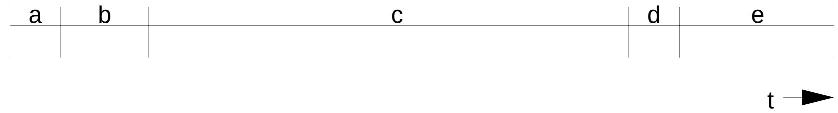

Figure 5-2: Overview of One-Way Delay Contribution

Notes to figure 5-2:

- a = transmit buffer time (time to configure an ADB out of p octets);
- b = space link processing latency at the transmitter side (time for VCDU handling, R-S coding, queuing to physical channel);
- c = signal travel time on the communication link;
- d = space link processing latency receiver side (time for VCDU handling, R-S decoding);
- e = receive buffer time (time for synchronization buffering (Doppler correction) and for restitution of isochronism to the output audio data).

This delay overview is in an analog way also valid for the video or still-image signal delay.

5.3.2 Audio Delay Budget

The delay budget of a space audio link using geostationary relay satellites is as follows:

1) **Transmit Buffer Time** (here: t_a, see figure 5-2)

duration: mainly depends on the number of octets (p) per ADB and the frequency of the audio data octets

t_a = p/f (audio data octets) (*)

(*) Assumption: m = n, which means only one ADF in an ADP.

2) **Space Link Processing Latency** (here: t_b+d, see figure 5-2)

duration: equivalent of 3 VCDU periods (in reference to the paper: *Analysis of Video and Audio Data Transport*, R. Carper, April 1989, reference [10])

t_b+d = ca. 3 VCDU periods
3 Mbps physical channel: t_b+d = 10.2 ms(*)
100 Mbps physical channel: t_b+d = 0.3 ms(*)

3) **Signal Travel Time on the Communication Link** (here: t_c in figure 5-2)

duration: varies between 240 and 290 ms for one direction; a round-trip delay in such a case may reach 580 ms

t_c = 240 ms ... 290 ms for 1 TDRS hop

4) **Receive Buffer Time** (here: t_e, see figure 5-2)

(due to Doppler shift/synchronization management)
duration: an example for the calculation of the Receive Buffer Time can be found in annex A.

t_e = 24 ms ... 47 ms

5.3.3 Multiplexing/Delay Relation Overview

A rough overview of the relationship between p, ADP frequency f(ADP)(**) and the resulting end-to-end delay for unidirectional audio data transmission is given in figure 5-3.

(*) VCDU/CVCDU period duration

The duration of a VCDU/CVCDU period depends on the physical channel data rate and the VCDU/CVCDU length.

Example: the VCDU/CVCDU length is 10200 bits

VCDU/CVCDU periods:

physical channel data rate	3 Mbps	100 Mbps
VCDU/CVCDU period	3.4 ms	0.1 ms

(**) Assumption: m=n, which means only one ADF in an ADP.

A short conclusion in relation to the audio service data multiplexing can be given:

When a short delay time is required, a small p-value must be chosen which implies a high ADP frequency.[(*)] If only one audio channel were concerned this would lead to short ADBs/ADFs/ADPs compared with the available capacity within a VCDU/CVCDU. A waste of bandwidth would be the result (excluding the Insert service). By taking into account the VCA service as a preferred CPN service, the audio channel multiplexing (more than one audio channel) will be appropriate to overtake the available bandwidth when a short delay time is the primary requirement.

In Annex B a recipe of the audio service data multiplexing configuration is given.

5.3.4 Audio Signal Doppler-Shift Resynchronization

For earth-orbiting spacecraft the data transmission via relay satellites is affected with a frequency variation due to the Doppler shift. To cope with that frequency shift (a positive as well as a negative shift) the recommendation for an audio data synchronization should be: to expand the receive buffer (at the receiver side of the transmission link) by a number of octets equivalent to the time base shift (as an additional time base shift buffer). The receive buffer readout will then be performed with the ground or onboard fixed clock for the audio data delivery.

Annex C gives an evaluation example and more detailed information about the audio signal resynchronization.

[(*)] Assumption: m=n, which means only one ADF in an ADP.

Assumptions for this example:

– VCA service is the applied CPN service;

– A2 Operational Audio service, uncompressed 64 kbps (8-bit/8-kHz audio data);

– a basic delay time estimation of 300 ms (latency due to space link processing latency, one-way signal travel time, and receive buffer time);

– p = octets per ADB in an ADF length of ADF ≤ size SDU(*) p(max) = 1265 octets (grade-3 CPN service) if n = 1 (1 ADB per ADF);

– f(ADP) = ADP frequency.(*)

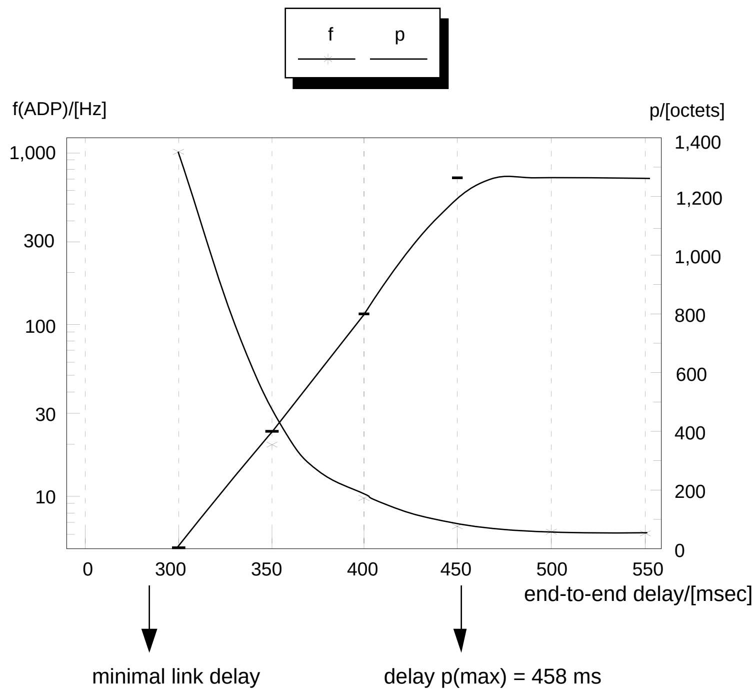

Figure 5-3: Example of Audio Service Data Multiplexing Parameter p and f(ADP)

(*) Assumption: m=n, which means only one ADF in an ADP.

5.4 BIDIRECTIONAL AUDIO COMMUNICATION ASPECT

Audio data can be unidirectionally transmitted only via the CPN or SLS. To establish a “quasi bidirectional” audio connection, two unidirectional audio channels (one for downlink, the other for uplink) have to be switched together at the chosen audio service access points onboard or on ground. For the implementation of the two separate audio channels of this “quasi bidirectional” audio connection, the user requirements and particularly the delay aspect have to be taken into account.

It should be considered that the round-trip delay shall be optimally maintained under 900 milliseconds. CCITT goals for link round-trip delay are set to be 600 milliseconds for a meaningful interactive conversation, but physical limits may require changes in operational constraints. The round-trip delay is here defined only for one geostationary hop. If more geostationary hops have to be considered, a larger signal delay occurs.

6 VIDEO TRANSPORT OVER THE CPN

This section defines the video data handling for the video data transmission via the CPN/SLS.

Subsection 6.1 describes the video source/output data structure, 6.2 defines the structure of the video service data stream, and 6.3 defines the applicable CPN/SLS services for the video service data transmission. Subsection 6.4 gives information about the recommended video service data multiplexing parameters.

6.1 VIDEO SOURCE/OUTPUT DATA STRUCTURE

The transmission of a video data stream with its temporal constraints (image-to-image interval) has to guarantee in any case the maintenance or reconstruction of the image interval isochronism at the receiver side.

Up to now, most video data sources have provided a fixed output data rate. In order to recreate the constant-rate video data stream while maintaining the temporal resolution of the image-to-image interval, the transmission system has been required to maintain end-to-end isochronism.

If video compression is applied, the compressor can generate fixed- or variable-rate output. Although the information content of the stream varies because of the variance in differences between adjacent frames, the output rate, in general, is kept constant in order to work over available fixed-rate commercial communications services and to maintain the constant frame rate. Therefore many compressed video sources also have bit-isochronism requirements.

Variable-rate video data can be generated by packet video codecs which take advantage of the variable information content of the frames generated by some compression algorithms. These systems do not place a strict isochronism requirement on the transmission channel at the bit level. The video packet rate will usually have requirements placed on it, however, in order to maintain a fixed frame rate at the receiver.

Because of the fixed output data rate of most current video sources, the CPN/SLS service used has to provide a sequence-preserving quasi-isochronous data transmission in order to get a continuous video data stream on the receiver side. To achieve a minimal delay and/or to provide several video channels within one video service data stream, a multiplexing scheme has to be implemented in analogy to the audio service multiplexing model.

6.2 VIDEO SERVICE DATA MULTIPLEXING MODEL

This section describes how to multiplex several video channels (of the same service type) within a video data stream.

This video multiplexing scheme at level 1 of the end-to-end video transport model (see figure 6-1) has to be applied:

– if a minimal video signal delay is considered;

– if more than one video channel within the video service data stream on one VC (maybe for an isochronous stereo application) must be transmitted.

Video services without these requirements can occupy the whole of a CCSDS VC without this multiplexing scheme. Packet video sources can occupy dedicated CCSDS Packet Channels and are not the concern of this section (see C.1.2 in reference [3]).

Two different data structures for video service data multiplexing are used:

– **Video Data Block**. The Video Data Block (VDB) data structure contains p contiguous octets of the same video channel.

– **Video Data Field**. The Video Data Field (VDF) data structure contains n VDBs, n being the number of video channels multiplexed in the video data stream. The n VDBs are arranged sequentially from VDB#1 to VDB#n within the VDF.

It is required that the n video channels being multiplexed within the VDF belong to the same video service. The values of n and p should be optimized together, taking into account:

– the type and bit rate of the CPN/SLS service used over the space link;

– the video channel bit rate;

– the required end-to-end delay for the video signals.

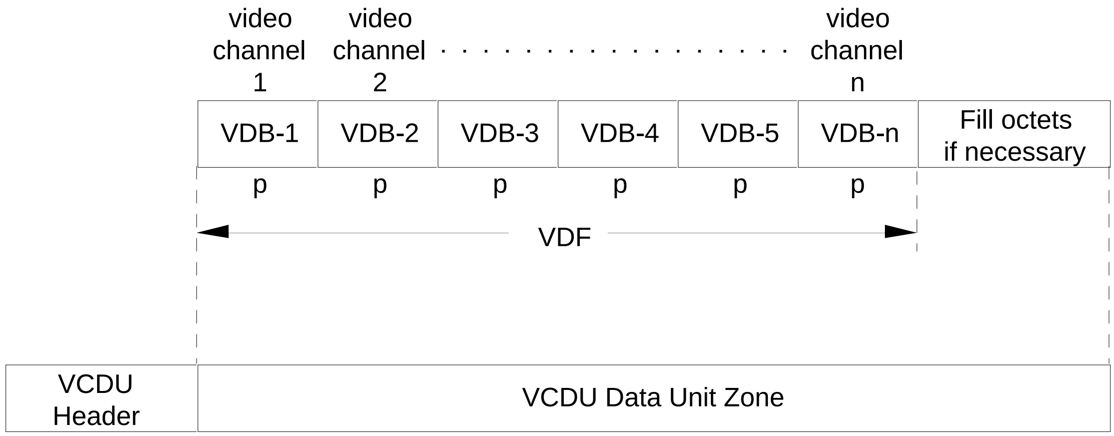

Figure 6-1: Video Service Data Multiplexing

6.2.1 Video Data Block

A VDB is composed of p contiguous octets from the same video channel. The video data might not be octet aligned at the output of the video coder. If it is not, octet alignment should be artificially created and preserved throughout the transmission over the space link by aligning video data octets with VDB octets.

The value of p has a considerable influence on the end-to-end video signal delay. The value of p and the assignment of video channels to VDB slots is not signaled in-line within the video formats. This information is supposed to be transmitted separately from the video data by the CPN management and signaling service.

The VDB is a fixed-length data structure of p octets. The p octets are all valid video data. This implies that, for a given video channel, VDBs are produced at a fixed rate of

F = (BR/8*p) blocks/sec (BR = video channel Bit Rate).

The VDF is a multiplex of n VDBs (n being fixed = number of multiplexed video channels). Therefore, VDFs are also produced at a fixed rate equal to F.

The overall rate of the CPN/SLS service used for the video data transmission has to match the aggregate video data rate precisely. This means that the VC output rate has to be driven by the video data rate and not by the link rate. Required therefore is a nonisochronous CPN/SLS service which, nevertheless, allows the reconstruction of the individual isochronous video data streams.

So VDFs/VDBs are buffered and transmitted asynchronously with respect to the fixed-rate F. The CPN/SLS service used should guarantee that each VDF shall not be delayed more than a given specification.

6.2.2 Video Data Field

The VDF is a fixed-length data structure consisting of n VDBs, n being the number of multiplexed video channels within the video data stream. n*p has to be smaller than or equal to the size of the SDU of the applied CPN/SLS service. The n VDBs are arranged in sequence within the VDF.

All the VDBs within a VDF represent a video channel of the same kind of video service (same parameters, same data rate, etc.). The number of octets per VDB, p, has to be the same for all VDBs within the VDF.

To prevent a jitter or delay of the incoming data out of the n multiplexed video channels in the VDF, the same sampling clock to generate the digital video service data octets should be used in all these channels.

The VDF will be inserted at the beginning of the SDU of the applied CPN/SLS service.

The value of n is not transmitted inline in the VDF structure. It is transmitted by the CPN management and signaling system.

6.3 CPN/SLS SERVICES FOR VIDEO

This subsection describes in detail the applicable CPN/SLS services to be used to transmit the data of the various video services. At a given time in the mission, each active video service generates its own video data streams. Each video data stream contains one or more video channels. Each video data stream will interface with a selected CPN/SLS service and be inserted in the form of VDFs to its SDUs.

Because of the large data rate of the video channels, in comparison with the audio channels, the CPN Insert service is not favored for the video service data transmission. Bitstream service (for nonmultiplexed video services data) and VCA service (for multiplexed video service data[*]) shall be the preferred CPN services.

6.3.1 Video on the Virtual Channel Access Service

The sequence-preserving VCA service can be used for video data stream transmission. Formatted VDFs are inserted into successive VCDUs or CVCDUs of the VCA service. The

[*] Possible also is a video data multiplexing structure carrying only the data out of one video channel.

structure of the VDFs/VDBs (i.e., values of p and n) is transmitted from sender side to receiver side by the CPN management and signaling system. This information is needed at both ends of the video link to establish communication.

The VDFs of a given video data stream will be transmitted in the VCA SDUs of a given VC.

Link access of the VCA SDU will be one of the following:

– Asynchronous, the video VC having a high priority, thus guaranteeing that the jitter and delay encountered by video will be kept relatively low. This mode of operation requires that the overall bit rate of the space link is relatively high.

– Synchronous, the video VC being sampled periodically by the VC multiplexer. This implies that the VCDU repetition rate is linked to the VDF repetition rate by an integer ratio, which in turn implies that the video system is synchronized with the space link transmission system. Therefore, this synchronous mode of operation should be avoided whenever possible.

The VDF should be inserted at the beginning of the VCA SDU (i.e., VCDU/CVCDU data field), starting from the first octet.

The length of the VDF submitted to the VCA service cannot be larger than the size of the VCA SDU (i.e., VCDU/CVCDU data field). If the length of the VDF is smaller than the size of the SDU, fill octets must be added after the VDF to fill in the SDU. Of course, the number of fill octets must be minimized to avoid losing bandwidth. The size of the VDF must be tailored to be as close as possible to the size of the VCA SDU (VCDU/CVCDU data field).

The VCA service should be the standard CPN service for video service data transmission over the space link.

6.3.2 Video on the Bitstream Service

The sequence-preserving Bitstream service is applicable for the transmission of video service data streams which are not multiplexed at level 1 of the end-to-end video transport model (see figure 6-1) (carrying only one video channel) and which have an embedded private synchronization scheme as a part of that video data.

The video data have only to be submitted to a CPN Bitstream service access point.

The Bitstream service should be the standard CPN service for the transmission of nonmultiplexed video data.

6.3.3 VIDEO ON THE ENCAPSULATION SERVICE

As an option the transfer of a fixed-rate video data stream via Encapsulation service is allowed, if the data traverse only the Space Link Subnet.

The Encapsulation service is a sequence-preserving service for the transmission of variable-length, octet-aligned data in an asynchronous mode. For data transmission the Encapsulation service takes a delimited user data unit and encapsulates it within CCSDS packets. These are multiplexed with other packets in the same packet channel and with packets from other packet channels into dedicated VCs.

The Encapsulation service may be used for the transmission of variable-rate video data such as that generated by packet video encoders. The Encapsulation protocol data units are multiplexed together in the CPN, and the user of an instance of the Encapsulation service will therefore be a nonmultiplexed source. The Encapsulation service is inherently asynchronous, although some time constraints may be placed on the bearer VC.

The use of a modification of the Encapsulation service with fixed-length packets and synchronous access to the VC has been put forward, but further work is required in this area.

For more information see Annex C of reference [3].

6.3.4 VIDEO ON THE PATH SERVICE

The Path service may be of interest for packet video sources which require flexibility of placement of the video codecs and of data transmission routing in the onboard and ground networks. The Path service provides internetworking using CCSDS packets as the protocol data unit. Like the Encapsulation service, transport of variable-length delimited user data structures is supported.

Care should be taken in the use of this service because of the unpredictable delay characteristics of the underlying onboard/ground subnets and because of their influence on the data sequencing. The Path service does not provide guaranteed in-sequence data over subnets, and sequencing can become corrupted. The video application may therefore need to apply its own sequencing mechanism. The CCSDS packet does provide a 14-bit packet sequence count field which may be used in this context. Alternatively, the Path service may generate a “Data Loss Indicator” where sequencing is violated.

For more information see Annex C of reference [3].

6.3.5 CPN GRADES OF SERVICE TO BE USED FOR VIDEO TRANSMISSION

The CPN provides three Grades of Service, which give three different levels of error-detection and correction capability. The error-handling functions are based on the fixed-length VCDU/CVCDU data structure transmitted over the space link. Grade of Service is a characteristic associated with a VC. Depending on the VC used to transmit a particular video SDU over the space link, the error protection will be different.

The grade-3 CPN service provides no error check or recovery of the user data (video data) within the VCDU. Video data transmission on a grade-3 VC minimizes the required bandwidth for this video VC (check bits of the R-S code are replaced by video data), but the BER encountered by the video data over the space link is higher. Therefore, grade-3 should be used for a particular video service if it is compatible with the end-to-end BER specified for this service (see annex D of reference [3]). The specified value for each service guarantees that the quality of video signal is not significantly degraded by the transmission errors.

The grade-2 CPN service provides an error-detection and correction function for each CVCDU based on an R-S code. In practice, this type of VC can be considered as providing error-free transmission over the space link. Using a grade-2 VC reduces the size of the VCDU data field because of added check bits of error code. Therefore, the bandwidth needed to transmit the same video data stream will be approximately 15-percent higher on a grade-2 VC compared to a grade-3 VC. As a result, it is recommended a that grade-2 VC be used for a given video service data transmission only if grade-3 does not meet the end-to-end BER requirement of this video service, or if, because of design constraints on the space link communication system, all the VCs are grade-2.

The grade-1 CPN service provides, in addition to the error-detection and correction function of the R-S code, an ARQ procedure, which guarantees (with a very high probability) the completeness and the correctness of the transmitted CVCDUs. Because of the retransmission protocol used, this service is not sequence preserving and is thus not acceptable for video data. Therefore, grade-1 CPN service should not be used for video.

In conclusion, only grade-2 and grade-3 sequence-preserving CPN services can be used for video service data transmission. The choice between the two should be made taking into account the end-to-end BER requirement of each video service as defined in Annex D of reference [3].

Detailed information about the CPN Grades of Service can be found in CCSDS AOS Blue and Green books (references [2] and [9]).

6.4 AUDIO/VIDEO SIGNAL RESYNCHRONIZATION

Synchronization of audio and video data means that the audio signal at the receiver side shall be maintained within the tolerances of 25 milliseconds leading video to 40 milliseconds lagging video (ANSI/EIA/TIA RS-250-C-1989). To achieve this two approaches can be introduced:

1. Transmission of the audio data and the video data in separate audio and video channels and synchronization of the two data streams at the receiver-side service access points by control and adaptation of the signal delays. The control of the delays can be achieved by comparing the time-tagging information in the audio and video data. The adaptation of the two signal delays can possibly be achieved by a regulated delay buffer for the audio data.

2. Transmission of the audio data as embedded audio within video data. For this there are two possible solutions:

   – The video coder provides the opportunity to multiplex time-synchronous audio data into the video data by an internal protocol/scheme (demultiplexing at the video decoder at the receiver side). If compressed audio data is embedded into the video data, the occurring delay due to the audio signal compression in relation to the video compression delay has to be considered for the synchronization of these two data steams.

   – The digital audio data and video data are multiplexed together or, as the case may be, demultiplexed in specific audio/video multiplexing devices (working, for example, in accordance to CCITT Rec. H221/H.242 Draft).

7 STILL-IMAGE TRANSPORT OVER THE CPN

This section defines the still-image data handling for the still-image data transmission via the CPN/SLS.

Subsection 7.1 describes the still-image source/output data structure, and 7.2 defines the applicable CPN/SLS services for the still-image service data transmission. Subsection 7.3 provides information about the interfaces between the still-image services and the CPN/SLS.

7.1 STILL-IMAGE SOURCE/OUTPUT DATA STRUCTURE

Digital still-image source data carries the image information represented by pels with fixed relation to a spatial position within the pictures. The still-image information is given by:

– the still-image color component (Y,U,V or RGB);

– the spatial resolution of the images;

– the radiometric resolution of the pels.

Unlike the video services, no temporal parameter has to be considered for the still-image services, because the still-image services are designed for delivering data of single images as isolated units.

A still-image data source submits only a determined volume of data per still-image to the CPN (if image data compression is used, the image data volume can be variable depending on the information content of the input image). Therefore, no data multiplexing model similar to those for audio and some video services is necessary for the still-image service. The data of a still-image is submitted as a packet to a CPN/SLS access point which uses an appropriate CPN/SLS service to transmit this type of packetized, noncontinuous data.

7.2 CPN/SLS SERVICES FOR STILL-IMAGES

This subsection describes the applicable CPN/SLS services to convey the still-image service data.

Because of the packetized data structure of the still-image data, the “Encapsulation service” and “Path service” are the appropriate CPN/SLS services for the still-image data transmission.

7.2.1 Still-Image Data on the Encapsulation Service

The Encapsulation service is a sequence-preserving service for the transmission for variable-length, octet-aligned data in an asynchronous mode. Encapsulation service can be used if the still-image data has only to traverse the Space Link Subnet.

The Encapsulation service takes a delimited user data volume and encapsulates it within CCSDS packets for subsequent multiplexing and transmission over the space link. If the data volume of a still-image exceeds 65536 octets, it has to be divided into data packets smaller than or equal to 65536 octets, because the Encapsulation service can only accept SDUs up to this size. This segmentation must be performed by the user still-image application.

7.2.2 Still-Image Data on the Path Service

The Path service is a non-sequence-preserving end-to-end service for the transmission of variable-length user data (moderate to large data volume) in an asynchronous mode. For still-image data there are fewer time constraints than for video data, so the Path service, although non–sequence preserving, can be used for the transmission of still-image data. The CCSDS packet incorporates a 14-bit sequence count which may be used by the still-image application for data reordering. Alternatively, the Path service may generate a “Data Loss Indicator” where sequencing is violated.

Path service has to be applied if the still-image data has to be transferred at the CPN Network layer.

The Path service should be the standard service for transmission of still-image data.

7.2.3 CPN Grades of Service to be used for Still-Image Transmission

The CPN provides three grades of error handling for user data to be transmitted (see also 6.3.5). The error handling is based on the fixed-length VCDU/CVCDUs transmitted on the physical data link.

The grade-3 CPN service is not acceptable for transmission of compressed still-image service data nor for scientific uncompressed still-image data.

The data delivered by a grade-2 transmission is practically error free at the receiver side. This is appropriate for compressed still-image data.

For still-image data transmission, the sequence-preserving grade-2 and, in the case of uncompressed still-image data (but with due reference to the quality requirements of the still-image data at the receiver), grade-3 CPN services are recommended. Detailed information about the CPN Grades of Service are given in the AOS Blue Book (reference [2]).

ANNEX A

EXAMPLE FOR THE CALCULATION OF THE RECEIVE BUFFER TIME

Example to Receive Buffer Time:

The delay time caused by the isochronism restitution might be neglected as the audio data octets can be taken out of the receive buffer directly after the ADP is been delivered. The main contribution to the delay time is the time related to the receive buffer extension which is used for the Doppler shift/synchronization management.

assumption: 30 minutes TDRS contact
worst case: one direction shift with delta_f_max

delta_f_max	=	$\pm$ f_0 * delta_v/c
	=	$\pm$ f_0 * 7.8 km/s/300*10^{3} km/s
	=	$\pm$ f_0 * 26*10^{-6}
delta_f_max/f_0	=	$\pm$ 26*10^{-6}
delta_f/f_0	$\leq$	$\pm$ 26*10^{-6}

Receive Buffer Time (t_e):

t_e(worst case)	=	delta_f_max/f_0 * TDRS contact time
t_e(worst case)	=	$\pm$ 26*10^{-6} * 1.8*10^{3} s
t_e(worst case)	=	$\pm$ 47 ms
t_e(typical)	=	t_e(worst case)/2
t_e(typical)	=	$\pm$ 24 ms

ANNEX B

A RECIPE OF THE AUDIO SERVICE DATA MULTIPLEXING CONFIGURATION

This annex recommends a configuration of the audio service data multiplexing for mapping to the CPN.

For audio data transmission the acceptable end-to-end delay will be an important requirement and has to be defined first for each audio service. For defining the accepted maximum end-to-end delay the operational aspects of the audio data transmission have to be considered (full duplex, unidirectional, short reaction time required . . .).

1. Depending on the accepted end-to-end delay the minimum theoretical required frequency of the ADPs "f(ADP)theo"(*) can be determined. If "f(ADP)theo" is in the order of the frequency of the CPN Protocol Data Units (PDUs) upon the physical data channel (ratio: (f(CPN-PDU physical channel)/f(ADP)) <2), the Insert service might be chosen for the audio data transmission.

   f(ADP)theo(*) = 1/t(transbuf)
   f(ADP)theo(*) = 1/(t(delay) - t(receivbuf) - t(travel) - t(slp_latency))

   definitions:

   – minimum ADP frequency (theoretical) = f(ADP)theo(*)

   – transmit buffer time = t(transbuf)

   – space link processing latency = t(slp_latency)

   – maximum acceptable end-to-end delay = t(delay)

   – receive buffer time = t(receivbuf)

   – signal travel time between transmitter and receiver = t(travel)

2. From the minimum theoretical ADP frequency "f(ADP)theo"(*) and the octet frequency of the voice data (e.g., 8 kHz), the theoretical number of audio data octets "p_theo" within an ADB can be evaluated. If the result of this evaluation is not an integer the next appropriate value of "p_theo" has to be chosen.

   p_theo = f(audio_octets)/f(ADP)theo(*)

(*) Assumption m=n, which means only one ADF per ADP.

definitions:

– theoretical number of octets in an ADB = p_theo

– audio data octet frequency (e.g., 8 k Hz) = f(audio_octets)

– minimum ADP frequency (theoretical) = f(ADP)theo(*)

3. If “p_theo” is larger than the number of octets of the SDU-size within a VCDU/CVCDU, a larger number of VCDU/CVCDUs is required. The minimum applicable SDU frequency f(SDU) for this case will be evaluated as follows:

f(SDU) = (p_theo/size(SDU)) * f(ADP)theo(*)

definitions:

– minimum SDU frequency = f(SDU)

– minimum ADP frequency (theoretical) = f(ADP)theo(*)

– theoretical number of octets in an ADB = p_theo

– number of octets of the SDU of the VCDU/CVCDU = size(SDU)

– minimum ADP frequency (usable) = f(ADP)use(*)

The real applicable value of ADP frequency “f(ADP)use” is the maximum of the values “f(ADP)theo” or “f(SDU)” evaluated above.

f(ADP)use(*) = MAXIMUM OF (f(ADP)theo(*); f(SDU))

An appropriate frequency “f(ADP)use” (equal to or larger than the evaluated value) has to be chosen.

Remark: The frequency of the CPN PDU upon the physical data channel f(CPN-PDU physical channel) can not be exceeded by “f(ADP)use” !

In the case of “f(SDU)” > “f(ADP)theo” the increased CPN-SDU frequency in relation to “f(ADP)theo” can be used for a reduction of the figure of p (the number of octets per ADB) in the ADF/ADP. This leads to a smaller ADP size (size ADP ≤ size SDU, so one SDU will carry one ADP) and a smaller audio signal delay as fixed at the beginning.

(*) Assumption m=n, which means only one ADF per ADP.

4. From the chosen ADP frequency "f(ADP)use" and the octet frequency of the audio data (e.g., 8 kHz) the practicable number of audio data octets p within 1 ADB can be evaluated. If the result of this evaluation is not an integer the next appropriate value of p has to be chosen.

   p = f(audio_octets)/f(ADP)use(*)

   definitions:

   – number of octets within an ADB = p

   – audio data octet frequency (e.g., 8 kHz) = f(audio_octets)

   – chosen ADP frequency = f(ADP)use(*)

5. The number of audio channels n transmitted with access to the same ADF/ADP can be evaluated from the value of p and the size of the SDU (size(SDU)) of the chosen CPN/SLS service.

   If the result of this evaluation is not an integer the next smaller integer as value for n can be chosen. Fill octets have to be included in the case of audio data transmission by VCA service to complete the SDUs.

   If Insert service is used, the number of audio channels n should be as small as possible.

   n = downrounded_integer(size(SDU)/p)

   definitions:

   – number of audio channels multiplexed within the same VC = n

   – number of octets within an ADB = p

   – number of octets of the SDU of the VCDU/CVCDU = size(SDU)

   – downrounded_integer: the evaluated value (size(SDU)/p) has to be decreased to an integer value

     examples:

```
downrounded_integer(1.000) = 1
downrounded_integer(1.001) = 1
downrounded_integer(1.999) = 1
downrounded_integer(2.000) = 2
```

(*) Assumption m=n, which means only one ADF per ADP.

ANNEX C

AUDIO SIGNAL RESYNCHRONIZATION WITH EVALUATION EXAMPLE

As evaluation example an (8-bit/8-kHz) audio channel is assumed.

For earth orbiting spacecraft the data transmission via relay satellites is affected with a frequency variation due to the Doppler shift of

delta_f/f_0 $\leq \pm 26*10^{-6}$ (see annex A)

In order to simplify the estimation of the time-base shift during a TDRS contact it might be assumed that in worst case 30 minutes have to be considered during which the Doppler effect accumulates to a maximum shift of:

t_rb(worst case) = delta_f_max/f_0 * TDRS contact time
t_rb(worst case) = $\pm 26*10^{-6} * 1.8*10^{3}$ s
t_rb(worst case) = $\pm$ 47 ms

t_rb : receive buffer time

The recommendation for the audio data synchronization should be, to expand the receive buffer by a number of octets equivalent to the time base shift (as an additional time base shift buffer).

size(tbs_buf)theo = t_rb(worst case) * audio octet freq.
size(tbs_buf)theo = $47*10^{-3}$ s * $8*10^{3}$ 1/s
size(tbs_buf)theo = 376 octets (for a 8-Bit/8-kHz audio channel)

size(tbs_buf)theo: theoretical size(time base shift buffer)
pro audio channel (8-bit/8-kHz)

For the practical implementation the evaluated value of size(tbs_buf)theo has to be enlarged to the next integer multiple (x) of the ADP size.(*)

size(tbs_buf)use = x * size(ADP)(*)

x : integer multiple of ADP size
size(ADP) : size(ADP) = m*n*p (here m = n = 1)
size(tbs_buf)use : practically used size(time base shift buffer)
pro audio channel (8-bit/8-kHz)

(*) Assumption m=n, which means only one ADF per ADP.

The receive buffer readout will be performed with the ground or onboard fixed clock for the audio octet delivery.

For resynchronization at a new TDRS contact point, the write/read pointers of the receive buffer have to be preset in accordance with the buffer extension value (size of time base shift buffer) in order to be able to cope with a positive as well as with a negative shift. Therefore the direction of the expected shift (positive or negative) has to be known.

size(rbuf_1_ch)min = size(ADP)(*) + size(tbs_buf)use

size(rbuf_1_ch)min : minimum size (receive buffer) pro audio channel
size(ADP) : size(ADP) = m*n*p (here m = n = 1)

in the case of audio channel multiplex:

size(rbuf)min = size(ADP)(*) + n * size(tbs_buf)use

size(rbuf)min : minimum size (receive buffer)
size(ADP) : size(ADP) = m*n*p (m=1)

(*) Assumption m=n, which means only one ADF per ADP.

ANNEX D

ACRONYMS AND ABBREVIATIONS

Term	Definition
A/V/SI	Audio, Video, And Still-Image
ADB	Audio Data Block
ADCT	Adaptive Discrete Cosine Transform
ADF	Audio Data Field
ADP	Audio Data Package
AOS	Advanced Orbiting Systems
BER	bit error rate
CCIR	International Radio Consultative Committee
CCITT	International Telegraph and Telephone Consultative Committee
CCSDS	Consultative Committee for Space Data Systems
CELP	code excited linear prediction
CMTT	Commission Mixte CCITT-CCIR pour les Transmissions Televisuelles et Sonores (joint CCITT-CCIR Commission for transmission of television and sound)
codec	coder/decoder pair
CPN	CCSDS Principal Network
CVCDU	Coded Virtual Channel Data Unit
DPCM	differential pulse code modulation
EIA	Electronic Industries Association
hi-fi	high fidelity
IEC	International Electrotechnical Commission
ISDN	Integrated Services Digital Network
JPEG	Joint Photographic Experts Group
kHz	kiloHertz
Mbps	megabits per second
min	minimum
PCM	pulse code modulation
PDU	Protocol Data Unit

Term	Definition
pel	picture element
R-S	Reed-Solomon
SDU	Service Data Unit
SLS	Space Link Subnetwork
VC	Virtual Channel
VCA	Virtual Channel Access
VCDU	Virtual Channel Data Unit
VDB	Video Data Block
VDF	Video Data Field